REMARKS

Claims 11-33 are currently pending in the subject application, and are presently under consideration. Claims 11-33 are rejected. Claims 11, 14, 16, 19, 21, 24, 28, and 31 have been amended. Favorable reconsideration of the application is requested in view of the amendments and comments herein.

I. Summary of Interview

Representative for Applicant thanks Examiner Armstrong for the opportunity to interview the present application on April 22, 2004. During the interview, representative for Applicant and Examiner Armstrong discussed claim 11 with regard to U.S. Patent No. 5,825,898 to Marash. It was believed that an understanding of the differences between claim 11 and U.S. Patent No. 5,825,898 to Marash was reached, but as Examiner Armstrong desired to confer with her Primary Examiner prior to finalizing her decision. In a voice mail message from Examiner Armstrong on May 26, 2004, and Examiner Armstrong indicated that after discussing the matter with her Primary Examiner that the rejections would be maintained.

II. Rejection of Claim(s) 11-13 and 16-18 Under 35 U.S.C. §103(a)

Claims 11-13 and 16-18 stand rejected under 35 U.S.C. §103(a) as being unpatentable over U.S. Patent No. 5,825,898 to Marash ("Marash") in view of U.S. Patent No. 4,581,758 to Coker et al. ("Coker"). Withdrawal of this rejection is respectfully requested for at least the following reasons.

Claim 11 has been amended to correct a grammatical error. Claim 11 recites a microphone array processing system for performance enhancement in noisy environments comprising a signal summation circuit for combining the filtered output signals from the microphones, whereby signal components resulting from the speech source combine coherently and signal components resulting from noise combine incoherently to produce an increased signal-to-noise ratio. The Office Action rejection relies on Marash for a teaching that a main channel matrix (Fig. 1, reference number 3) produces a weighted sum of outputs which filters a

signal coming in all directions to produce a signal coming in a specific direction. The Office Action then asserts that the main channel matrix of Marash reads on the signal summation circuit of claim 11. Applicant respectfully traverses this assertion.

The main channel matrix of Marash generates a main channel signal representing signals received in the direction of a source and contains both a source signal component and an interference signal component (col. 4, ll. 59-62). The main channel matrix generates the main channel as a weighted sum of outputs from a group of multipliers (col. 6, ll. 8-10). The filter weights can be any combination of fractions as long as their sum is equal to 1 (col. 6, ll. 10-11). Accordingly, the main channel signal disclosed in Marash is a combination of the voice and interference (i.e. noise) components from all of the microphones, but because of the fractional weighting in the combining of signals implemented by the main channel matrix, the main channel signal is no greater in amplitude than any one of the microphone signals individually. Therefore, the signal summation circuit of claim 11 is not taught by the main channel matrix of Marash because the signal components resulting from the speech source (i.e. voice component) do not combine coherently to produce an increased signal-to-noise ratio, as recited in claim 11. Instead, the weights applied by the main channel matrix would not provide an increased signal-to-noise ratio. Specifically, in order for the system in Marash to achieve an improved signal-to-noise ratio, it must subtract out the noise components by a difference unit (Fig. 1, reference number 8). Therefore, the signal output from the main channel matrix does not produce an increased signal-to-noise ratio, as recited in claim 11. The signal summation circuit of claim 11 is thus not taught by the main channel matrix of Marash.

In the Response to Arguments section (page 9) of the Office Action dated March 1, 2004, the signal summation circuit of claim 11 is also likened to summation block 260 of Marash (which is the same as the difference unit 8 of Fig. 1). It is respectfully submitted that the difference unit of Marash does not teach the signal summation circuit of claim 11. Marash teaches that the difference unit subtracts canceling signals from the delayed main channel to generate a digital output signal (col. 5, ll. 10-12). The canceling signals are generated by the adaptive filtering of the reference channels (col. 5, ll. 1-7), which represent signals received from

directions other than that of the signal source. Thus, the reference channels of Marash represent interference signals (i.e. noise components), and hence do not contain voice components (col. 4, ll. 63-67). Because there are no voice components in any of the signals output from the adaptive filters in Marash, there cannot be any coherently combining of signal components resulting from a speech source AND incoherently combining of signal components from noise, as recited in claim 11. Accordingly, the difference unit of Marash does not teach or suggest the signal summation circuit, as recited in claim 11.

In addition to the deficiency of the difference unit standing alone to teach the signal summation circuit of claim 11, claim 11 also recites a plurality of adaptive filters, one for each of the data microphones, for aligning each data microphone output signal with the output signal from the reference microphone. It is this alignment of data microphone output signals with reference microphone output signals that allows the signal components resulting from the speech source to combine coherently (Specification; page 11, line 29 - page 12, line 2). The adaptive filters of Marash generate canceling signals that are subtracted from the main signal to generate an output signal substantially free from an interference signal component because the canceling signals closely track the interference signal components (col. 8, ll. 47-54). Furthermore, the main channel signal of Marash is not an output from a reference microphone, as recited in claim 11. Instead, the main channel signal of Marash is a signal created from the fractional weighted sum of all output signals from the microphones, so long as the sum of such signals is one (col. 6, ll. 1-55). Accordingly, because Marash teaches that the interference signals are subtracted from the main channel signal (the fractional weighted sum of all output signals) to reduce the interference signal component, the adaptive filters are aligning the interference (i.e. noise) components of the main signal and interference signals. With such an alignment occurring in the adaptive filters of Marash, the canceling signals are then subtracted from the main channel signal to remove noise from the main channel signal and thus increase signal-to-noise ratio. This noise cancellation technique taught by Marash is not, however, fails to suggest the approach recited in claim 11; namely, coherently combining the voice components and incoherently combining the

noise components to increase signal-to-noise ratio. Therefore, the difference unit of Marash further does not teach the signal summation circuit of claim 11.

The addition of Coker does not cure the above-noted deficiencies of Marash with respect to claim 11. Coker is relied upon in the Office Action to show a teaching of a plurality of bandpass filters for eliminating from the microphone output signals a known spectral band containing noise, as recited in claim 11. Marash and Coker, however, taken alone or in combination, fail to teach or suggest the recitations of claim 11. Withdrawal of the rejection of claim 11, as well as claims 12 and 13 which depend therefrom, is respectfully requested.

Claim 16 has been amended to correct a grammatical error. Claim 16 recites a method of improving detection of speech signals, the method comprising adaptively filtering the microphone output signals in a plurality of adaptive filters, one for each of the data microphones, and thereby aligning each data microphone output signal with the output signal from the reference microphone; and combining the adaptively filtered output signals from the microphones in a signal summation circuit, whereby signal components resulting from the speech source combine coherently and signal components resulting from noise combine incoherently, to produce an increased signal-to-noise ratio. Accordingly, for substantially the same reasons described above with regard to claim 11, claim 16 is patentable over Marash, alone or in combination with Coker. Withdrawal of the rejection of claim 16, as well as claims 17 and 18 which depend therefrom, is respectfully requested.

For the reasons described above, claims 11-13 and 16-18 are patentable over the cited art. Accordingly, withdrawal of this rejection is respectfully requested.

III. Rejection of Claim(s) 14-15 and 19-20 Under 35 U.S.C. §103(a)

Claims 14-15 and 19-20 stand rejected under 35 U.S.C. §103(a) as being unpatentable over Marash in view of Coker and in further view of the Digital Signal Processing Handbook (1998) ("DSP Handbook"). Withdrawal of this rejection is respectfully requested for at least the following reasons.

Claim 14 has been amended to correct grammatical errors. Claims 14 and 15 depend from claim 11. Accordingly, reconsideration and allowance of claims 14 and 15 are respectfully requested.

Claim 19 has been amended to correct a grammatical error. Claims 19 and 20 depend from claim 16, which, as described above, should be allowed over the cited art. Therefore, reconsideration and allowance of claims 19 and 20 are respectfully requested.

For the reasons described above, claims 14-15 and 19-20 are patentable over the cited art. Accordingly, withdrawal of this rejection is respectfully requested.

IV. Rejection of Claim(s) 21-33 Under 35 U.S.C. §103(a)

Claims 21-33 stand rejected under 35 U.S.C. §103(a) as being unpatentable over Marash in view of the DSP Handbook. Withdrawal of this rejection is respectfully requested for at least the following reasons.

Claim 21 has been amended to clarify that it is the updated filter weight value being converted from the frequency domain to the time domain. Amended claim 21 recites a method for improving detection of speech signals. However, similar to the above description with regard to claim 11, the adaptive filtering process of Marash does not align adaptively filtered interference signals with reference microphone data, as recited in claim 21. Instead, in Marash, adaptively filtered interference signals are aligned with a main channel signal, which main channel signal is created from the fractional weighted sums of all of the microphones in the system. Marash further teaches that the main channel signal is input to an inhibitor, which in turn outputs an inhibit signal to the adaptive filters to perform the adaptive filtering process (col. 5, ll. 15-18). Therefore, Marash does not teach or suggest the step of converting a reference microphone data to a frequency domain, nor does it teach or suggest updating the filter weight value with the reference microphone data.

The addition of the DSP Handbook does not cure the deficiencies of Marash in failing to teach what is set forth in claim 21. The DSP Handbook teaches a general adaptive filtering algorithm for increasing convergence speed. Referring to page 22-17, figure 22.11 of the DSP

Handbook, the bottom right of the figure shows the desired response signal d_k (analogous to the main channel signal of Marash) input to an adder, which adds a positive value of the desired response signal d_k to a negative value of the filter output vector y_k to create an error signal e_k. It is this error signal e_k that is converted to the frequency domain using a FFT operation (bottom right of figure 22.11) and that is input to the adaptive filter to update the filter with current weight values. Neither the DSP Handbook nor Marash, taken individually or in combination, teaches or suggests converting reference microphone data to the frequency domain, as recited in claim 21. In particular, the desired response signal (presumably main channel signal of Marash, which is provided to the adaptive filter) is not converted to the frequency domain in the DSP Handbook. Instead, the DSP Handbook teaches that it is the difference of the desired response signal and the filter output vector that is converted to the frequency domain. Thus, any adaptive filtering performed based on the combined teachings of Marash and the DSP Handbook, as suggested in the Office Action, would be performed based on the main channel signal of Marash (the only signal provided to the adaptive filter), such that no updating of filter weight values can be made with reference microphone data, as recited in claim 21.

In addition, the Office Action suggests that the DSP Handbook teaches updating the filter weight value with the reference microphone data, as recited in claim 21. A closer reading of the DSP Handbook and Marash demonstrates the weakness of this suggestion. In figure 22.11, the DSP Handbook depicts a block for updating a weight vector W_k, but is silent as to the basis for performing the update. As described above, Marash fails to teach or suggest the existence of a reference microphone data that is utilized to adaptively filter. Instead, Marash teaches the application of the main channel signal to the adaptive filter, which main channel signal is a weighted sum of all microphone signals. Accordingly, it is submitted that neither the DSP Handbook nor Marash, either individually or in combination, teach or suggest any updating of a filter weight value with a reference microphone data, as recited in claim 21.

Representative for Applicant respectfully requests that Examiner identify with specificity the nature of this updating portion of the adaptive filtering process to support the assertion that the combination of references teaches updating of the filter weight value with the reference

microphone data. In particular, an explanation as to how, absent improper hindsight, one of ordinary skill in the art would modify Marash to update the filter weight value, as recited in claim 21.

Because Marash does not teach or suggest the use of a reference microphone data in its method of adaptive filtering, and because the DSP handbook does not teach or suggest converting a reference microphone data to a frequency domain as well as updating the filter weight value with the reference microphone data, claim 21 is patentable over Marash, alone or in any combination with the DSP Handbook and/or Coker. For these reasons, the rejection of claim 21, as well as claims 22-27 which depend therefrom, is respectfully requested to be withdrawn.

Additionally, claim 24 has been amended to correct a typographical error. Claim 24 recites that the updating filter weight value comprises W(k+1) = W(k)+μ(Ref(k) – X(k))*conj(Y) where k is the data block number and μ is a small adaptive step constant. The DSP Handbook teaches an equation for calculating the updated filter weight in the frequency domain (page 22-17, equation 22.33) of $W_{k+1} = W_k + \mu X_k^H * E_k$, wherein $E_k = FFT(d_k - y_k)$. In this analysis, it is assumed that the variable X (upper and lower case) in the DSP Handbook corresponds to the variable Y (upper and lower case) in claim 24, and vice verse. It is further assumed that, as appearing on page 22-15 (last paragraph) of the DSP Handbook, capital letters are used to denote the frequency-domain variables and lowercase letters to denote the time-based variables. Rewriting equation 22.33, wherein Ref(k) of claim 24 is assumed to correspond to the frequency domain conversion of d_k, and conj(Y) is assumed to correspond to X_k^H, and substituting E_k as well as X and Y, $W_{k+1} = W_k + \mu(FFT(d_k - y_k)) * conj(X)$. This equation is different from that shown in claim 24, the difference being inherent to the difference in adaptive filtering technique, as described above with regard to claim 21. Because the Fourier Transform is a complex form of the Fourier integral, the term $FFT(d_k - y_k)$ of rewritten equation 22.33 is not equal to the term (Ref(k) – X(k)) of claim 24. Rewritten 22.33 is converting the difference of the time domain terms to the frequency domain, instead of subtracting two frequency domain terms. Applicant submits, that these differences are a result of the different approaches taught by the DSP

handbook and that recited in claim 24. Withdrawal of the rejection and allowance of claim 24 are respectfully requested.

Claim 28 has been amended to clarify that it is the updated filter weight value being converted from the frequency domain to the time domain. Amended claim 28 recites a system for detecting speech signals comprising a means for converting a reference microphone data to the frequency domain and a means for updating the filter weight value with the reference microphone data. For the reasons described above with regard to claim 21, claim 28 is also patentable over Marash, alone or in combination with the DSP Handbook and /or Coker. Withdrawal of the rejection of claim 28, as well as claims 29-33 which depend therefrom, is respectfully requested.

Claim 31 also has been amended to correct a typographical error. Claim 31 recites that the updating filter weight value comprises $W(k+1) = W(k)+\mu(Ref(k) - X(k))*conj(Y)$ where k is the data block number and μ is a small adaptive step constant. For the same reasons described above with regard to claim 24, claim 31 is patentable over the Marash, alone or in combination with the DSP Handbook and/or Coker. Withdrawal of the rejection and allowance of claim 31 is respectfully requested.

For the reasons described above, claims 21-33 are patentable over the cited art, and their allowance is respectfully requested.

V. CONCLUSION

In view of the foregoing remarks, Applicant submits that the present application is in condition for allowance and respectfully requests reconsideration and allowance of this application.

The Examiner is invited to call the undersigned if a telephone interview would be helpful to further prosecution of this matter.

Please charge any deficiency or credit any overpayment in the fees for this amendment to our Deposit Account No. 20-0090.

Respectfully submitted,

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